JC19 Rec'd PCT/PTO 01 JUN 2001

FORM PTO-1390 (REV 11-2000) DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE		ATTORNEY'S DOCKET NO. 851663.429USPC
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371		U.S. APPLICATION NO. (If known, see37 CFR 1.5) **Unknown** 09/857120
INTERNATIONAL APPLICATION NO. **PCT/SG98/00098**	INTERNATIONAL FILING DATE **02 December 1998 (02.12.1998)**	PRIORITY DATE CLAIMED **02 December 1998 (02.12.1998)**

TITLE OF INVENTION

FIXED-POINT MULTIPLICATION FOR ADPCM SPEECH CODER

APPLICANT(S) FOR DO/EO/US

LEONG, Foo, Yuen

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)). The submission must include items (5), (6), (9) and (21) indicated below.
4. ☒ The US has been elected by the expiration of 19 months from the priority date (Article 31).
5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2)).
   a. ☒ is attached hereto (required only if not communicated by the International Bureau).
   b. ☐ has been communicated by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☐ An English language translation of the International Application as filed (35 U.S.C. 371(c)(2)).
   a. ☐ is attached hereto
   b. ☐ has been previously submitted under 35 U.S.C. 154(d)(4).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3)).
   a. ☐ are attached hereto (required only if not communicated by the International Bureau).
   b. ☐ have been communicated by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☒ have not been made and will not be made.
8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)).
10. ☐ A English language translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 11 to 20 below concern document(s) or information included:

11. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
13. ☐ A FIRST preliminary amendment.
14. ☐ A SECOND or SUBSEQUENT preliminary amendment.
15. ☐ A substitute specification.
16. ☐ A change of power of attorney and/or address letter.
17. ☐ A computer-readable form of the sequence listing in accordance with PCT Rule 13ter.2 and 35 U.S.C. 1.821 – 1.825.
18. ☒ A second copy of the published international application under 35 U.S.C. 154(d)(4)
19. ☐ A second copy of the English language translation of the international application under 35 U.S.C. 154(d)(4).
20. ☐ Other items of information:

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U.S. APPLICATION NO. (If known, see 37 CFR 1 5) **Unknown** 09/857120	INTERNATIONAL APPLICATION NO. **PCT/SG98/00098**	ATTORNEY'S DOCKET NUMBER **851663.429USPC**

21. ☒ The following fees are submitted:

Basic National Fee (37 CFR 1.492(a)(1)-(5)):

CALCULATIONS PTO USE ONLY

Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO and International Search Report not prepared by the EPO or JPO $1000.00

International preliminary examination fee (37 CFR 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO $860.00

International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO $710.00

International preliminary examination fee (37 CFR 1.482) paid to USPTO but all claims did not satisfy provisions of PCT Article 33(1)-(4)..................... $690.00

International preliminary examination fee (37 CFR 1.482) paid to USPTO and all claims satisfied provisions of PCT Article 33(1)-(4) $100.00

ENTER APPROPRIATE BASIC FEE AMOUNT =				$860.00	
Surcharge of $130.00 for furnishing the oath or declaration later than ☐ 20 ☒ 30 months from the earliest claimed priority date (37 CFR 1.492(e)).				$130.00	
Claims	Number Filed	Number Extra	Rate		
Total Claims	4 - 20 =	0	x $ 18.00	$.00	
Independent Claims	2 - 3 =	0	x $ 80.00	$.00	
Multiple dependent claim(s) (if applicable)			+ $270.00	$.00	
TOTAL OF ABOVE CALCULATIONS =				$990.00	
☐ Applicant claims small entity status. See 37 CFR 1.27. The fees indicated above are reduced by 1/2.				$.00	
SUBTOTAL =				$990.00	
Processing fee of $130.00 for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(f)). +				$.00	
TOTAL NATIONAL FEE =				$990.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). $40.00 per property +				$.00	
TOTAL FEES ENCLOSED =				**$990.00**	
				Amount to be refunded:	
				charged	

a. ☒ A check in the amount of $990.00 cover the above fees is enclosed.

b. ☐ Please charge my Deposit Account No. in the amount of $_ to cover the above fees. A duplicate copy of this sheet is enclosed.

c. ☒ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. **19-1090**. A duplicate copy of this sheet is enclosed.

d. ☐ Fees are to be charged to a credit card. WARNING: Information on this form may become public. Credit card information should not be included on this form. Provide credit card information and authorization on PTO-2038.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

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46,274
REGISTRATION NUMBER

EXPRESS MAIL NO. EL755730825US

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant	:	Foo Yuen Leong
Application No.	:	09/857,120
Filed	:	June 1, 2001
For	:	FIXED-POINT MULTIPLICATION FOR ADPCM SPEECH CODER

Docket No.	:	851663.429USPC
Date	:	July 27, 2001

Box Non-Fee Amendment
Commissioner for Patents
Washington, DC 20231

PRELIMINARY AMENDMENT

Commissioner for Patents:

Prior to an examination on the merits, please amend the above-identified application as follows:

In the Claims:

Please amend claim 1 as follows:

1. (Amended) A method for encoding speech or voice band data by way of adaptive differential pulse coded modulation including an adaptive predictor procedure which implements an adaptive predictive filter for generating a signal estimate from quantized difference signal values, reconstructed signal values and respective predictor coefficients according to a predetermined multiplication and accumulation operation, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, including performing multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed

difference signal values, reconstructed signal values and respective predictor coefficients according to a predetermined multiplication and accumulation operation, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, including performing multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values, summing the double word length fixed point partial product values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

Please add new claims 5-11 to read as follows:

5. (New) An audio encoder for adaptive differential pulse coded modulation of a sequence of digital audio samples, comprising:

a pulse code modulator (PCM) converter to accept the sequence of digital audio samples and convert the samples to a uniform pulse code modulated data sequence;

an adder coupled to the PCM converter and receiving the uniform pulse code modulated data sequence therefrom, the adder generating a difference signal indicating a difference between a selected one of the uniform pulse code modulated data sequence and a signal estimate of prior ones of the uniform pulse code modulated data sequence;

an adaptive quantizer to assign a quantized value of a predetermined number of binary digits to indicate an adaptive quantized value of the difference signal to thereby form an output signal;

an inverse adaptive quantizer to generate a quantized difference signal from the output signal;

an adaptive predictor to generate the signal estimate from input quantized difference signal values, input reconstructed signal values and respective predetermined predictor coefficients wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values;

a fixed point multiplier to perform multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values; and

an accumulator to sum the double word length fixed point partial product values to form a double word length predictor sum.

6. (New) The encoder of claim 5 wherein the accumulator rounds the predictor sum to a single word length fixed point representation of the signal estimate, the signal estimate being provided to the adder.

7. (New) The encoder of claim 5 wherein said single word length representation comprises 16 bits and said double word length representation comprises 32 bits.

8. (New) The encoder of claim 5 wherein the adaptive predictor comprises a first processor block to receive the reconstructed signal in a first format and to convert the reconstructed signal to a second fixed point format.

9. (New) The encoder of claim 8 wherein the first format is a signed magnitude format and the second format is a twos-complement format.

10. (New) The encoder of claim 8 wherein the fixed point multiplier performs multiplication in fixed point format on data provided in the second fixed point format.

11. (New) The encoder of claim 8 wherein the fixed point multiplier performs multiplication in a twos-complement format.

REMARKS

Claim 1 has been amended to correct two typographical errors. New claims 5-11 are added.

The Applicants request examination of the amended claims. If questions arise regarding this case, the Examiner is invited to contact the undersigned at (206) 622-4900.

Respectfully submitted,

LEONG, Foo, Yuen

Seed Intellectual Property Law Group PLLC

Michael J. Donohue

Michael J. Donohue
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VERSION WITH MARKINGS TO SHOW CHANGES MADE

In the Claims:

Claim 1 has been amended as follows:

1. (Amended) A method for encoding speech or voice band data by way of adaptive differential pulse coded modulation including an adaptive predictor procedure which implements an adaptive predictive filter for generating a signal estimate from quantized difference signal values, reconstructed signal values and respective predictor coefficients according to a predetermined multiplication and accumulation operation, wherein the quantized difference signal values and reconstructed signal values are represented by ~~signal~~ single word length fixed point binary values, including performing multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values, summing the double word length fixed point partial product ~~valuesto~~ values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

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FIXED-POINT MULTIPLICATION FOR ADPCM SPEECH CODER

Field of the Invention

This invention relates to the implementation of a digital speech coder for the transmission of speech or voice band data over a communications network.

Background of the Invention

In order to transmit speech or voice band data over a communications network in a digital form, one of the methods that may be used to encode the input data for transmission is Adaptive Differential Pulse Coded Modulation (ADPCM). The ADPCM algorithm achieves speech compression by combining adaptive quantization and differential PCM. Adaptive quantization adjusts the step size of the quantizer as the signal changes. This allows the algorithm to accommodate variations in the signal amplitude. Differential PCM involves transmitting the difference between the current and previous signal sample instead of simply transmitting the current sample itself. The difference signal obtained in this way tends to have a much lower dynamic range compared to the original signal and may therefore be quantized to a specific signal-to-noise ratio with fewer bits.

In practice, the difference signal is computed from the current signal sample and a signal estimate determined by an adaptive predictor. The adaptive predictor uses signal estimates of previous samples to obtain an approximation of the current sample. This is performed in both the encoder and decoder so that they are synchronised with each other and there will not be any accumulation of errors in the reconstructed signal at the decoder output.

In ITU-T Recommendation G.726, the adaptive predictor is represented by a two-pole, six-zero adaptive predictive filter. The combination of poles and zeros enables the filter to deal with any general input signal. The sixth-order all-zero filter is needed to stabilise the filter and prevent it from drifting into oscillation. The filter coefficients are updated based on a

simplified gradient algorithm.

The signal estimate is computed by:

$$s_e(k) = \sum_{i=1}^{2} a_i(k-1)s_r(k-i) + \sum_{i=1}^{6} b_i(k-1)d_q(k-i)$$

where s_e : signal estimate
s_r : reconstructed signal
d_q : quantized difference signal
a_i, b_i : predictor coefficients

The range of values of the predictor coefficients is limited to ±2 and are stored as 16-bit fixed point values. The quantized difference signal and reconstructed signal can vary between -32768 to 32767. Initially 16-bit fixed point values, they are then converted to floating point and stored. The aforementioned ITU-T recommendation specifies that the multiplication operation should be performed in floating point, by converting all inputs to floating point values with 6 bits of mantissa and 4 bits of exponent. The resulting product is then converted back into a 16-bit fixed point number.

Summary of the Invention

In accordance with the present invention, there is provided a method for encoding speech or voice band data by way of adaptive differential pulse coded modulation including an adaptive predictor procedure which implements an adaptive predictive filter for generating a signal estimate from quantized difference signal values, reconstructed signal values and respective predictor coefficients according to a predetermined multiplication and accumulation operation, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, including performing multiplication in fixed point format between the respective said predictor coefficients and the quantized difference

signal values and reconstructed signal values to generate respective double word length fixed point partial product values, summing the double word length fixed point partial product values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

The present invention also provides An adaptive differential pulse coded modulation encoder for encoding speech or voice band data for transmission over a communications network, including an adaptive predictor having an adaptive predictive filter for generating a signal estimate from input quantized difference signal values, input reconstructed signal values and respective predetermined predictor coefficients, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, the adaptive predictive filter including a multiplier which performs multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values, and an accumulator for summing the double word length fixed point partial product values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

Preferably the single word length representations comprise 16 bit binary values and the double word length representations comprise 32 bit binary values. However, it will be appreciated that other length words are possible within the scope of the invention, depending upon the type of computational processing equipment the invention is to be implemented on.

The invention is described in greater detail hereinafter, by way of example only, through description of a preferred embodiment thereof and with reference to the accompanying drawing which illustrates a generalised block diagram of an ADPCM encoder.

Detailed Description of the Preferred Embodiment

The present invention relates to adaptive differential pulse coded modulation (ADPCM) of speech or voice band data for transmission over a communications network, of the type which is described in ITU-T Recommendation G.726, the disclosure of which is incorporated herein by reference. The ADPCM encoder in the ITU-T recommendation converts a 64 kbit/s PCM input into an ADPCM compressed output for transmission. The accompanying drawing figure illustrates a block diagram of an ADPCM encoder according to the ITU-T recommendation. Referring to the figure, an A-law or μ-law PCM input stream is first converted to uniform PCM. A difference signal is then obtained by subtracting an estimate of the input signal from the input signal itself. An adaptive quantizer is used to assign a quantized value of a predetermined number of binary digits to the value of the difference signal for transmission to the decoder. An inverse adaptive quantizer is arranged to produce a quantized difference signal from the quantized value output from the adaptive quantizer. The input signal estimate is added to the quantized difference signal to produce the reconstructed version of the input signal. Both the reconstructed signal and the quantized difference signal are operated upon by an adaptive predictor which produces the input signal estimate, thus forming a feedback loop.

The embodiment of the invention herein described is concerned primarily with the adaptive predictor portion of the ADPCM encoder, and in particular the filtering operation of the adaptive predictor. Because of the floating point multiplications, the filtering operation of the adaptive predictor is the most complex block of the ADPCM algorithm. According to the ITU-T recommendation, this involves first converting the fixed point inputs to floating point, multiplying the mantissas and adding the exponents, and finally converting the floating point product back to fixed point representation. In addition to the computational complexity of this operation, when the values d_q and s_r are close to the 16-bit limit, converting them to floating point will result in a loss of precision even before the multiply operation can take place. This is because only 6 bits are retained for the mantissa of the floating point number.

The preferred embodiment of the present invention provides a way in which to perform the operations more efficiently and accurately.

Since all the input values are originally available in 16-bit fixed point format, it is possible to perform the multiplication directly in fixed point. This eliminates the need to convert the values between fixed and floating point formats. To reduce the errors due to loss of precision, the full 32-bit intermediate products are kept during accumulation. At the end of the filter operation, the final accumulated product is then rounded off to 16 bits.

The preferred embodiment of the invention primarily involves four function blocks that are defined in ITU-T Recommendation G.726, namely the FLOATA, FLOATB, FMULT and ACCUM blocks. The functions of these blocks as utilised in the ITU-T recommendation are described briefly below with reference to the figure and the signal estimate equation mentioned above.

FLOATA: This block receives the quantized difference signal d_q as input, where the quantity d_q is defined as a 15 or 16 bit signed binary magnitude. The quantized difference signal d_q is converted into a floating point value. This is performed by computing the exponent and mantissa and combining the sign bit, 4 exponent bits and 6 mantissa bits into one 11 bit word.

FLOATB: This block receives the reconstructed signal s_r as input, where the quantity s_r is defined as a 16 bit twos-complement quantity. The reconstructed signal s_r is converted into a floating point value. This is performed by computing the exponent and mantissa and combining the sign bit, 4 exponent bits and 6 mantissa bits into one 11 bit word.

FMULT: This block multiplies predictor coefficients with the corresponding quantized difference signal or reconstructed signal. The multiplication is done in a floating point format, and thus the predictor coefficients, which are defined as

16 bit twos-complement quantities, are first converted into floating point representations. The products of the multiplication operations are signal estimate partial products (WAn, WBn), which are also defined as 16 bit twos-complement quantities, requiring a conversion from the floating point multiplication result.

ACCUM: This block operates on the signal estimate partial products to perform the summing portion of the operation represented by the equation discussed above. The partial products of the signal estimates (WA1, WA2, WB1, WB2, WB3, WB4, WB5 and WB6) are received as input and summed to obtain the complete signal estimate s_e. All of the quantities are twos-complement representations.

The preferred embodiment employs fixed point multiplication rather than floating point computation, which requires a number of modifications as described below. Replacing the floating point multiplication with a fixed point multiplication eliminates the need to convert values between fixed and floating point formats. This significantly reduces the complexity of the overall algorithm. By omitting the fixed-to-floating point conversion, the full precision of the original values is preserved, thereby reducing errors due to loss of precision. As a result, an improvement in the quality of the decoded signal can be achieved. The primary modifications to the ITU-T recommended ADPCM adaptive predictive filter which are implemented in the preferred embodiment are summarised below.

The FLOATA block ordinarily takes the reconstructed signal and converts it from 16-bit signed magnitude format to floating point. This block is re-defined to convert the signed magnitude numbers to 16-bit twos-complement numbers instead. The FLOATB block ordinarily takes the signal estimate and converts it from 16-bit twos-complement to floating point. This block is no longer needed and is discarded from the system.

The FMULT block normally performs several functions, namely converting the predictor

filter coefficients from 16-bit twos-complement to floating point, performing floating point multiplication by adding the exponents and multiplying the mantissas and finally converting the product back into a 16-bit twos-complement number. The preferred embodiment requires that all these functions be discarded and replaced by a simple fixed-point multiplication which multiplies two 16-bit twos-complement numbers to give a 32-bit product. The full 32 bits of the result is retained. No truncation to 16 bits is performed.

The ACCUM block ordinarily adds the 16-bit predictor outputs together to form the signal estimate. The preferred embodiment requires that the accumulation function be modified to operate on 32-bit inputs. After the final accumulation, the result is then rounded off to 16 bits to give the signal estimate.

Thus, the preferred embodiment of the invention requires the ability to perform 16x16 bit fixed-point multiplication and to store the 32-bit result for subsequent arithmetic operations. The following procedures, in pseudocode, implement the preferred embodiment by redefining the FLOATA, FLOATB, FMULT and ACCUM blocks originally specified in ITU-T G.726. The modified procedures in combination retain the functionality of the ITU-T recommendation, although not in strict compliance with the specification. Table 1, below, provides a description of the format of the variables used in the procedures.

Name	Bits	Binary representation	Description
A1, A2	16 TC	S, 0, ...,-14	Delayed second order predictor coefficients
B1, ..., B6	16 TC	S, 0, ...,-14	Delayed sixth order predictor coefficients
DQ	16 SM	S, 14, ..., 0	Quantized difference signal
DQ0	16 TC	S, 14, ..., 0	Quantized difference signal with delay 0
DQS	1 TC	S	Sign bit of quantized difference signal
SE	15 TC	S, 13, ..., 0	Signal estimate
SEZ	15 TC	S, 13, ..., 0	Sixth order predictor partial signal estimate
SR	16 TC	S, 14, ..., 0	Reconstructed signal
SR0	16 TC	S, 14, ..., 0	Reconstructed signal with delay 0
SR1, SR2	16 TC	S, 14, ..., 0	Reconstructed signal with delays 1 and 2
WA1, WA2	32 TC	S, 16, ..., -14	Partial product of signal estimate
WB1,..,WB6	32 TC	S, 16, ..., -14	Partial product of signal estimate

TC denotes twos-complement representation

SM denotes signed magnitude representation

S denotes sign bit

Table 1 Format and description of variables

Procedure 1: FLOATA

Function: Convert 16-bit signed magnitude to 16-bit two's complement

```
DQS = DQ >> 15                  | Get the sign bit.
DQM = DQ & 32767                | Compute magnitude.
if DQS = 1, DQ0 = DQM           | Convert magnitude to
else DQ0 = -DQM                 | twos-complement.
```

Procedure 2: FLOATB

Function: Copy 16-bit twos-complement number from input to output

SR0 = SR

Procedure 3: FMULT

Function: Multiply predictor coefficients with corresponding quantized difference signal or reconstructed signal. Multiplication is done in fixed point format.

WAn = An x SRn	Perform fixed point
WBn = Bn x DQn	multiplication.

Procedure 4: ACCUM:

Function: Addition of predictor outputs to form the partial signal estimate (from the sixth order predictor) and the signal estimate.

SEZI = WB1 + WB2 + WB3 + WB4 + WB5 + WB6	Sum for partial signal estimate.
SEI = SEZI + WA2 + WA1	Complete sum for signal estimate.

SEZ = SEZ1 > > 14

SE = SEI > > 14

The foregoing detailed description of the preferred embodiment of the invention has been presented by way of example only, and is not intended to be considered limiting to the invention as defined in the claims appended hereto.

Claims:

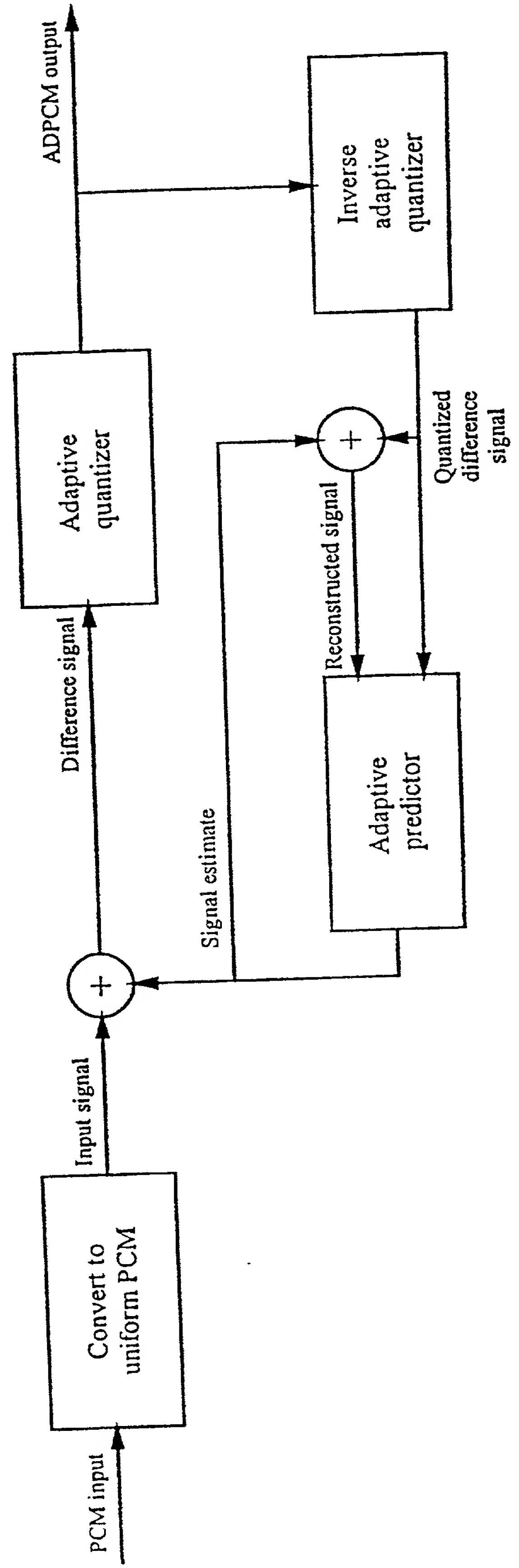
1. A method for encoding speech or voice band data by way of adaptive differential pulse coded modulation including an adaptive predictor procedure which implements an adaptive predictive filter for generating a signal estimate from quantized difference signal values, reconstructed signal values and respective predictor coefficients according to a predetermined multiplication and accumulation operation, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, including performing multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values, summing the double word length fixed point partial product values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

2. A method as claimed in claim 1, wherein a said single word length representation comprises 16 bits and a said double word length representation comprises 32 bits.

3. An adaptive differential pulse coded modulation encoder for encoding speech or voice band data for transmission over a communications network, including an adaptive predictor having an adaptive predictive filter for generating a signal estimate from input quantized difference signal values, input reconstructed signal values and respective predetermined predictor coefficients, wherein the quantized difference signal values and reconstructed signal values are represented by single word length fixed point binary values, the adaptive predictive filter including a multiplier which performs multiplication in fixed point format between the respective said predictor coefficients and the quantized difference signal values and reconstructed signal values to generate respective double word length fixed point partial product values, and an accumulator for summing the double word length fixed point partial product values to form a double word length predictor sum and rounding the predictor sum to a single word length fixed point representation of said signal estimate.

4. An adaptive differential pulse coded modulation encoder as claimed in claim 3, wherein a said single word length representation comprises 16 bits and a said double word length representation comprises 32 bits.

09/857120



ADPCM Encoder

DECLARATION AND POWER OF ATTORNEY

As the below-named inventor, I declare that:

My residence, post office address, and citizenship are as stated below under my name.

I believe I am the original, first, and sole inventor of the invention entitled "FIXED-POINT MULTIPLICATION FOR ADPCM SPEECH CODER," which is described and claimed in the specification and claims of International Patent Application No. PCT/SG98/00098, which was filed on 2 December 1998 and for which a patent is sought.

I have reviewed and understand the contents of the above-identified specification and claims, as amended by any amendment specifically referred to herein (if any). I acknowledge my duty to disclose information of which I am aware which is material to the patentability and examination of this application in accordance with 37 C.F.R. § 1.56(a).

I hereby claim foreign priority benefits under 35 U.S.C. § 119 of the foreign patent application listed below:

PRIOR FOREIGN/PCT APPLICATION(S) AND ANY PRIORITY CLAIMS UNDER 35 U.S.C. 119:			
COUNTRY	APPLICATION NUMBER	DATE OF FILING	PRIORITY CLAIMED UNDER 35 USC 119
PCT	PCT/SG98/00098	2 December 1998	Yes

I hereby appoint DAVID V. CARLSON, Registration No. 31,153; MICHAEL J. DONOHUE, Reg. No. 35,859; ROBERT IANNUCCI, Reg. No. ~~33,514;~~ E. RUSSELL TARLETON, Reg. No. 31,800; ERIC J. GASH, Reg. No. ~~46,274;~~ KEVIN S. COSTANZA, Registration No. 37,801; SUSAN D. BETCHER, Reg. No. 43,498; BRIAN L. JOHNSON, Registration No. 40,033; GEORGE C. RONDEAU, JR., Reg. No. ~~28,893;~~ BRIAN G. BODINE, Reg. No. 40,520; CHARLES J. RUPNICK, Reg. No. 43,068; TIMOTHY L. BOLLER, Reg. No. 47,435; and FRANK ABRAMONTE, Reg. No. ~~38,066~~; comprising the firm of Seed Intellectual Property Law Group PLLC, 701 Fifth Avenue, Suite 6300, Seattle, Washington 98104-7092; and THEODORE E. GALANTHAY, Registration No. 24,122; LISA K. JORGENSON, Registration No. ~~34,845;~~ ROBERT D. McCUTCHEON, Registration No. 38,717; MARIO DONATO, Reg. No. 37,816 ~~and~~ NAINESH SHAH, Reg. No. 40,166; as my attorneys to prosecute this application and transact all business in the

Patent and Trademark Office connected therewith. Please direct all telephone calls to Eric J. Gash at (206) 622-4900 and telecopies to (206) 682-6031.

I further declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Y. Y. Foo

Foo Yuen Leong

Date 21 June 2001

Residence	:	City of Singapore, Country of Singapore.
Citizenship	:	Singapore
P.O. Address	:	Block 102 Commonwealth Crescent, #10-108 Singapore 140102

183007_1 DOC